EAST WEST UNIVERSITY

Department of EEE

Section: 01

Course Code: EEE309

Course Name: Digital Signal Processing

Course Instructor's Name: Dr. Halima Begum,
Assistant Professor, EEE, EWU

Project

Submission Date: 25 December,2023

Group No: 04

Submission By:

01. Arpon Podder(2020-1-80-005)
02. Rohit Bhowmick(2020-1-80-006)
03. Kazi Mashrur Rahman (2020-1-80-057)
04. MD. Arif Mollah (2020-1-80-093)

PROJECT TITLE: Digital Filter Design

OBJECTIVE

The objective of this project is to design a digital filter which can suppress the noise in ECG data due to power line interference. As we know ECG data consists of different kinds of noises like electrode contact noise, instrumentation noise and electrosurgical noise etc. Our goal will be to minimize implementation cost and consider implementation using fixed precision algorithms.

INTRODUCTION

One of the easiest and quickest methods for assessing the heart is an electrocardiogram, or ECG. We can determine the degree of physiological stimulation someone is feeling by using a technology that gathers electrical impulses produced by their heartbeat. However, many disturbances, including power line interference, electrode contact noise, instruments noise, electrosurgical noise, etc, frequently tamper with ECG data. The goal of this project is to create an appropriate filter in order to mitigate power line interference.

COURSE OUTCOME

The course outcomes that we achieve throughout the projects are:

CO#4: Design filters subject to different specifications and constraints.

CO#5: Investigate issues related to signal processing by designing and conducting and data analysis.

Observation of the Given Signal

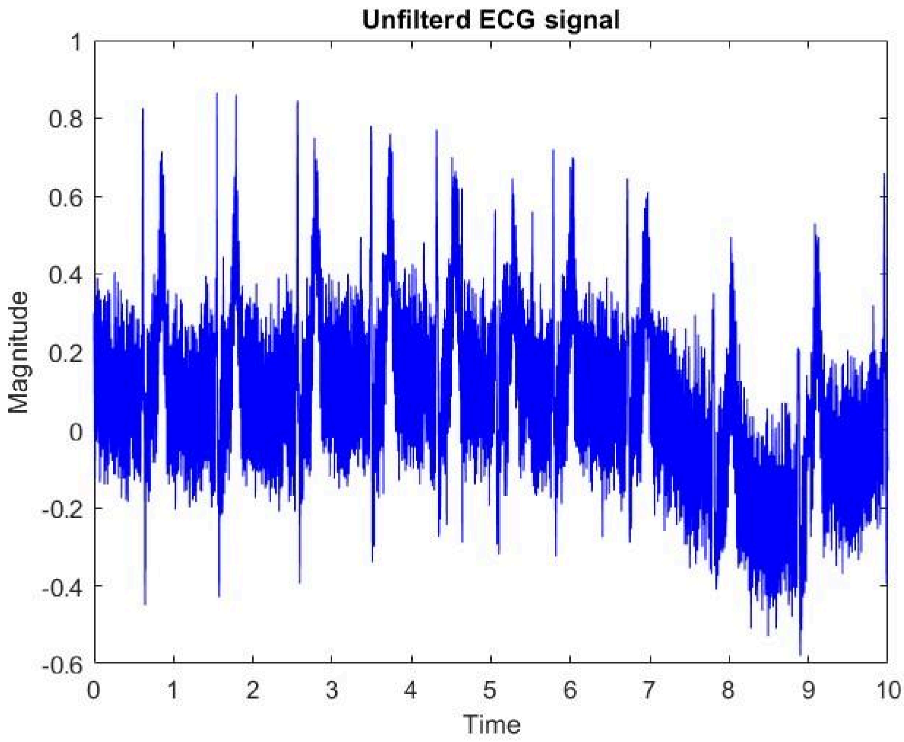

Figure 01: Corrupted ECG Signal in Time Domain.

Comment: We can clearly see from Figure: 01 clearly that there is so much noise present in the signal.

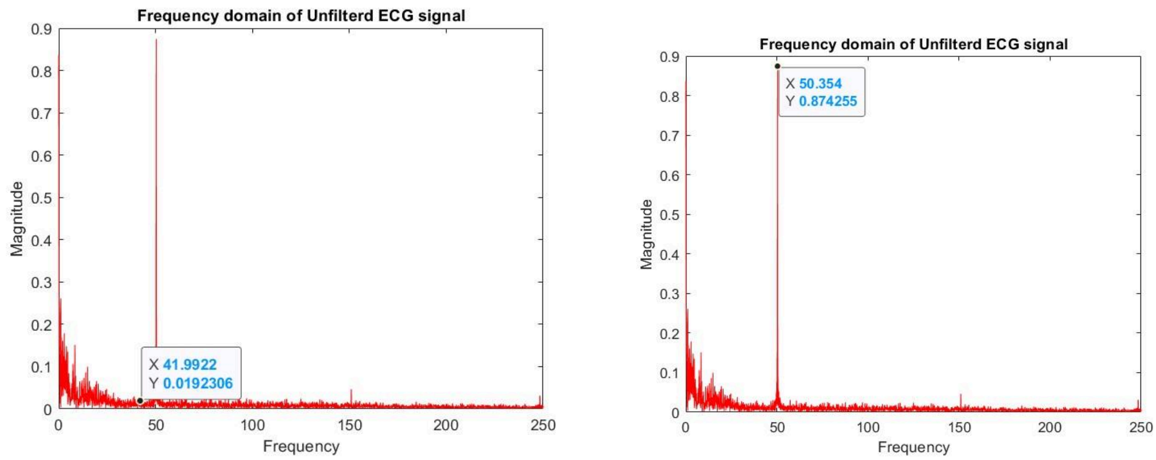

Figure-02: Frequency component analysis of corrupted ECG data.

Comment: We have got a spike at frequency 50 Hz. From the plot above we can see that the ECG value magnitude gradually decreases and the amplitude remains zero after approximately 41 Hz. So the ECG information can be found between 0 Hz to 41 Hz according to the plot.

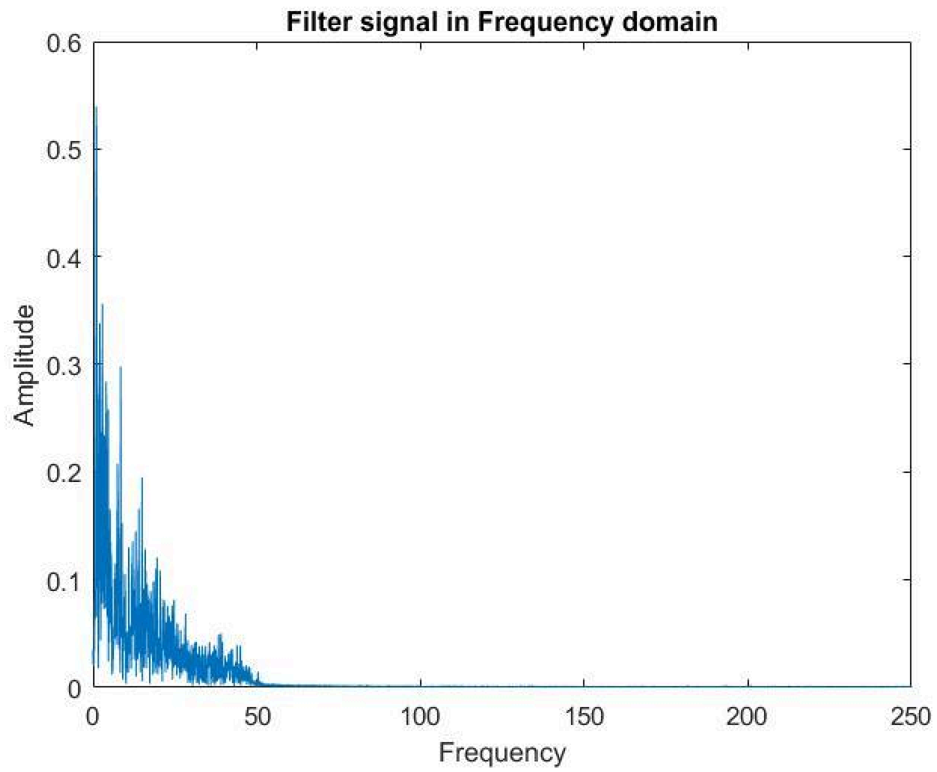

Figure-04: Provided filtered ECG signal (Frequency Domain)

Filter selection:

Selection of Filter:

We need to design a filter that will be able to pass frequencies from 0 to 41 Hz and stop higher frequency components to remove noise (due to power line interference, 50.4 Hz). In this case, we can choose the FIR lowpass filter or IIR lowpass filter. So we have chosen the FIR lowpass filter for our project.

Reason for Choosing FIR Filter:

The electrical signal from the heart is recorded in medical testing using ECG signals. Thus, in this instance, a linear phase response is crucial. Although an IIR filter will change the phase angle, it will save cost since it takes less memory. There are no linear phase characteristics offered by the IIR filter. While maintaining a linear phase response in the data, the FIR filter uses more memory than the IIR filter. Thus, a linear phase response may be obtained by employing this filter.

FILTER DESIGN:

PARAMETERS

Let,
Sampling Frequency, F_S = 500 Hz
Passband Edge Frequency, F_P = 41 Hz
Stopband Edge Frequency, F_{S1} = 50 Hz

Cutoff Frequency, $F_C = \frac{F_P + F_S}{2}$

$= \frac{50+41}{2} = 45.5$ Hz

Transition Width, $\Delta F = 9\ HZ$
$\Delta\omega = 2\pi\Delta F = \frac{2\times\pi\times9}{500} = 0.036\pi$

Let,
Passband Ripple, $\delta_P = 0.015$
Stopband Ripple, $\delta_S = 0.001$ [Stopband ripple lower than passband ripple]
$\delta = min[\delta_P, \delta_S] = 0.001$
We know,
Stopband Attenuation, $A = -20log_{10}(\delta) = -20log_{10}(0.001)$
$= 60$ dB

As, A = 60 dB which is greater than 53 dB. So, we can use only "Blackman Window" & "Kaiser Window".

KAISER WINDOW CALCULATION

We know,
Ripple Control Parameter, $\beta = 0.1102(A - 8.7) = 5.65326$
Window Length & Order,
$M + 1 = [1 + \frac{A-8}{2.28\Delta\omega}]$
$= [1 + \frac{60-8}{2.28\times0.036\pi}]$
$= 202.22 \approx 203$
$\therefore$ M = 202

$\therefore$ length, M + 1 = 203
$\therefore$ Order, M = 202

BLACKMAN WINDOW CALCULATION

We know,

$$\Delta f = \frac{5.5}{M+1}$$

=> $\frac{9}{500} = \frac{5.5}{M+1}$

$\therefore$ M + 1 = 305.556 ≈ 306
$\therefore$ length, M + 1 = 306
$\therefore$ Order, M = 305

From the calculation of **"Kaiser Window"** & **"Blackman Window"**, we can see that the length is higher in **"Blackman Window"**. That's why we selected Kaiser Window for our filter design for reducing memory.

Filter Implementation (MATLAB)

The filter is designed according to our filter calculation.

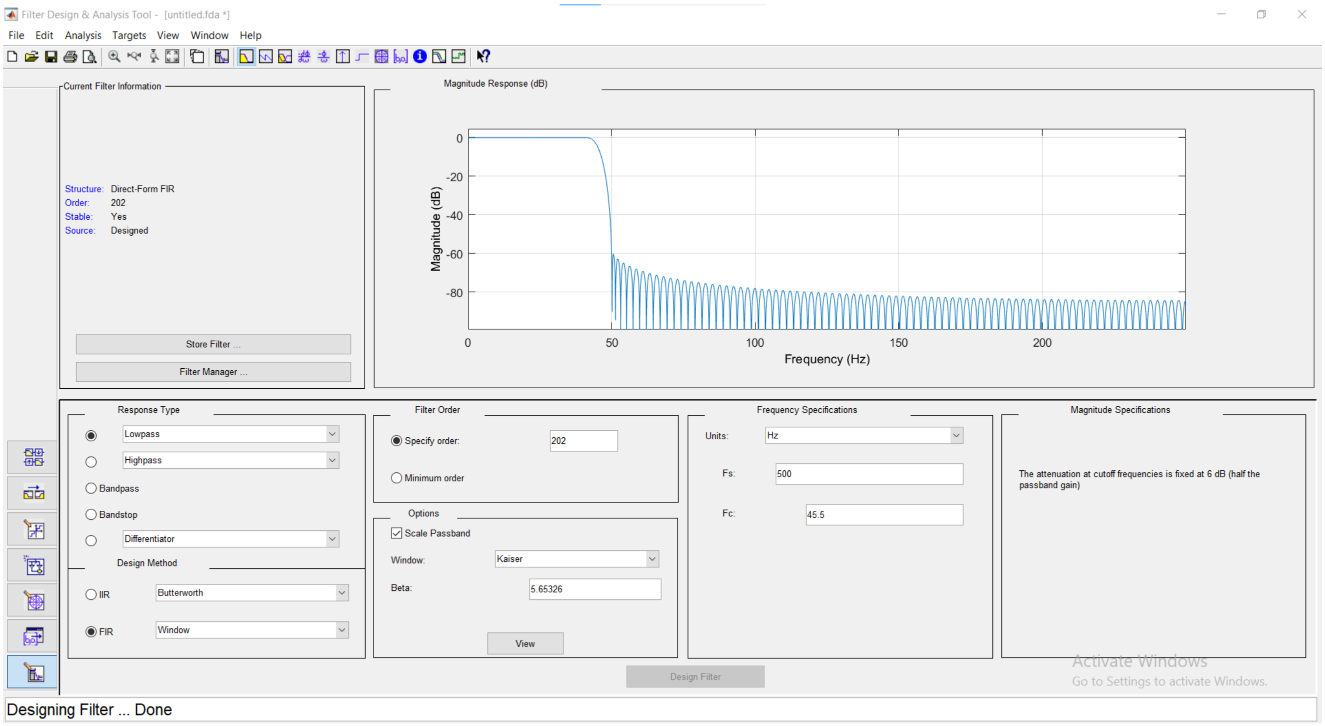

Figure-05: Magnitude Response of Designed Filter

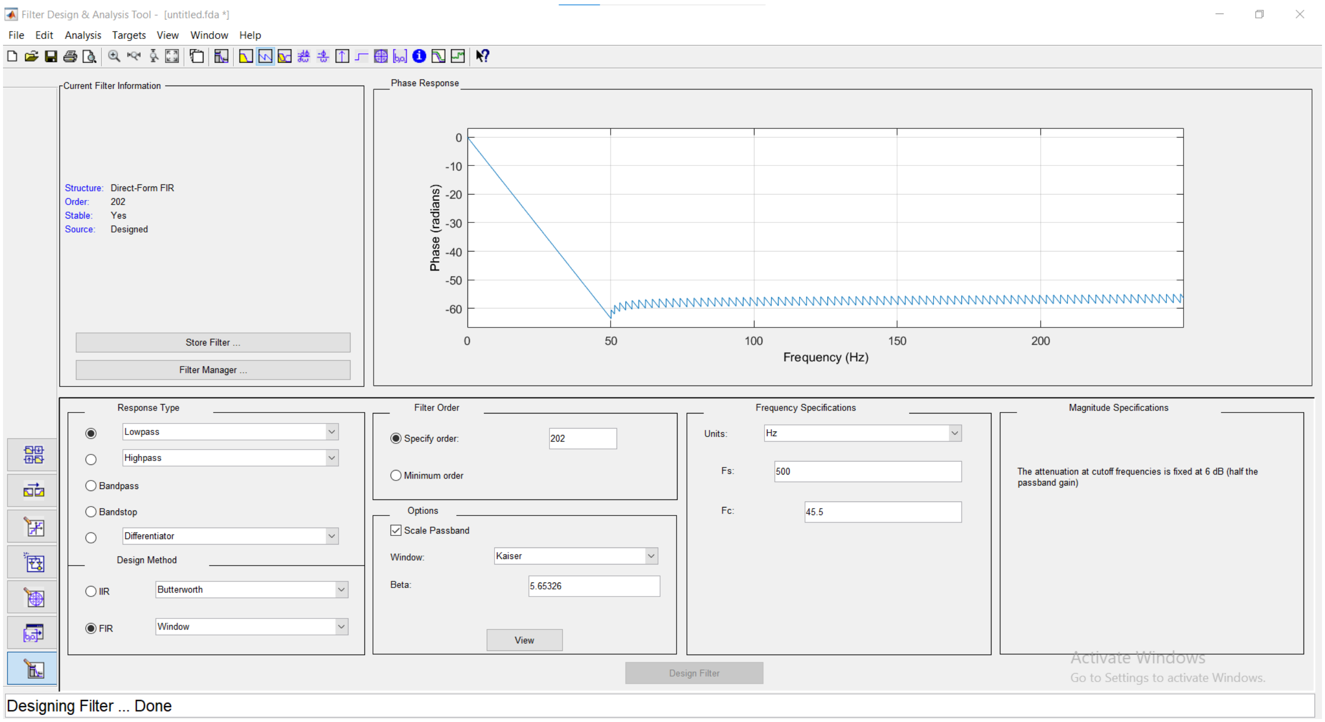

Figure-06: Phase Response of Designed Filter

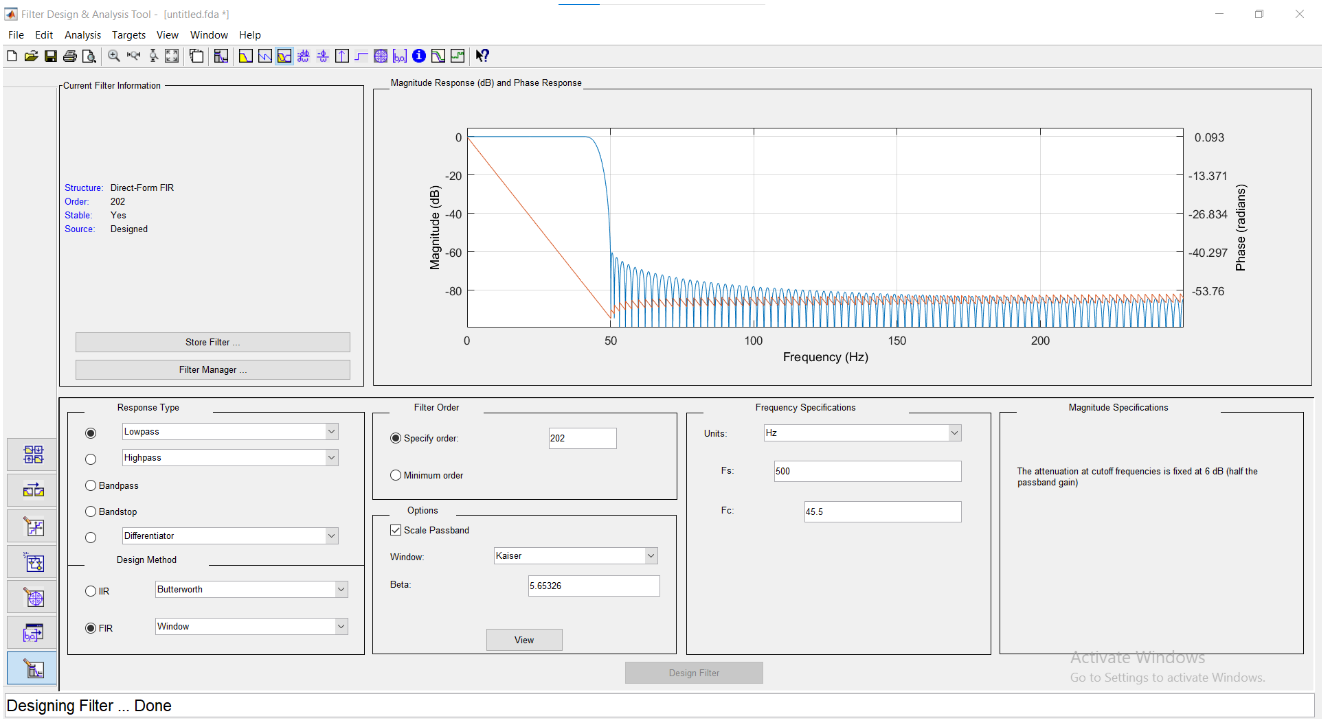

Figure-07: Magnitude & Phase Response Together of Designed Filter

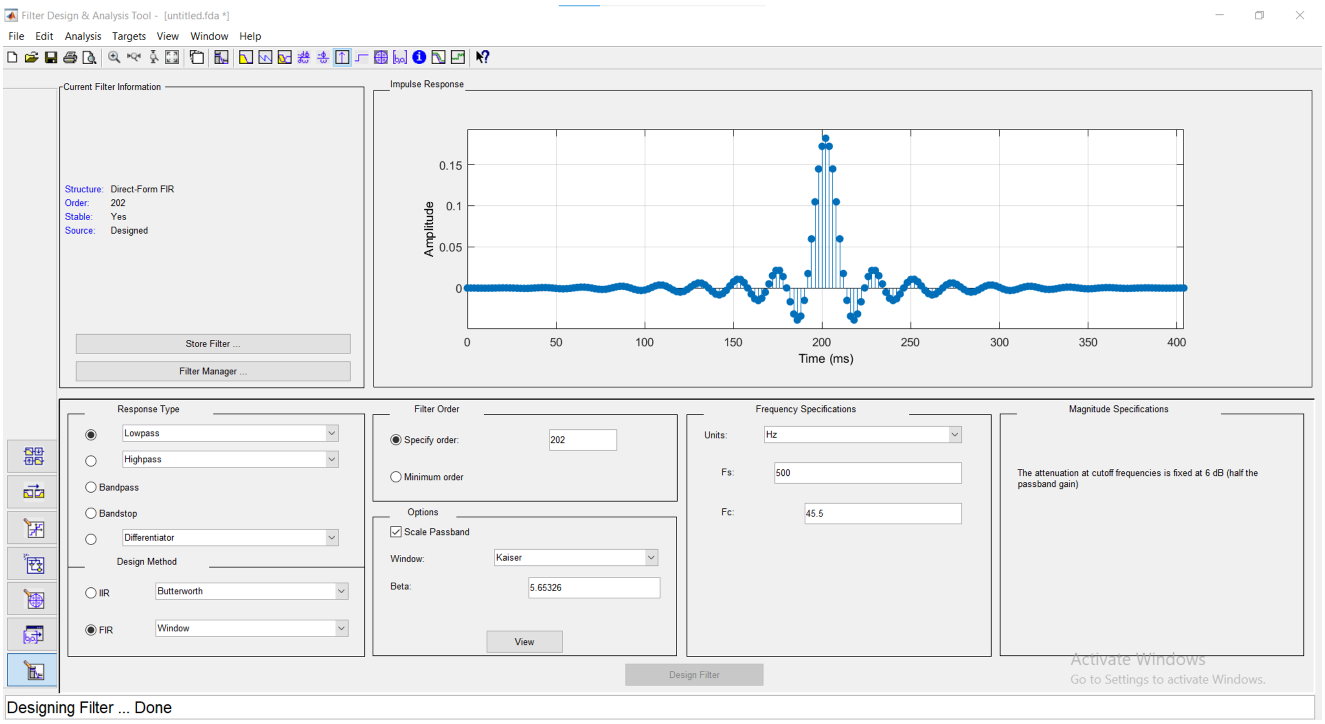

Figure-08: Impulse Response of Designed Filter

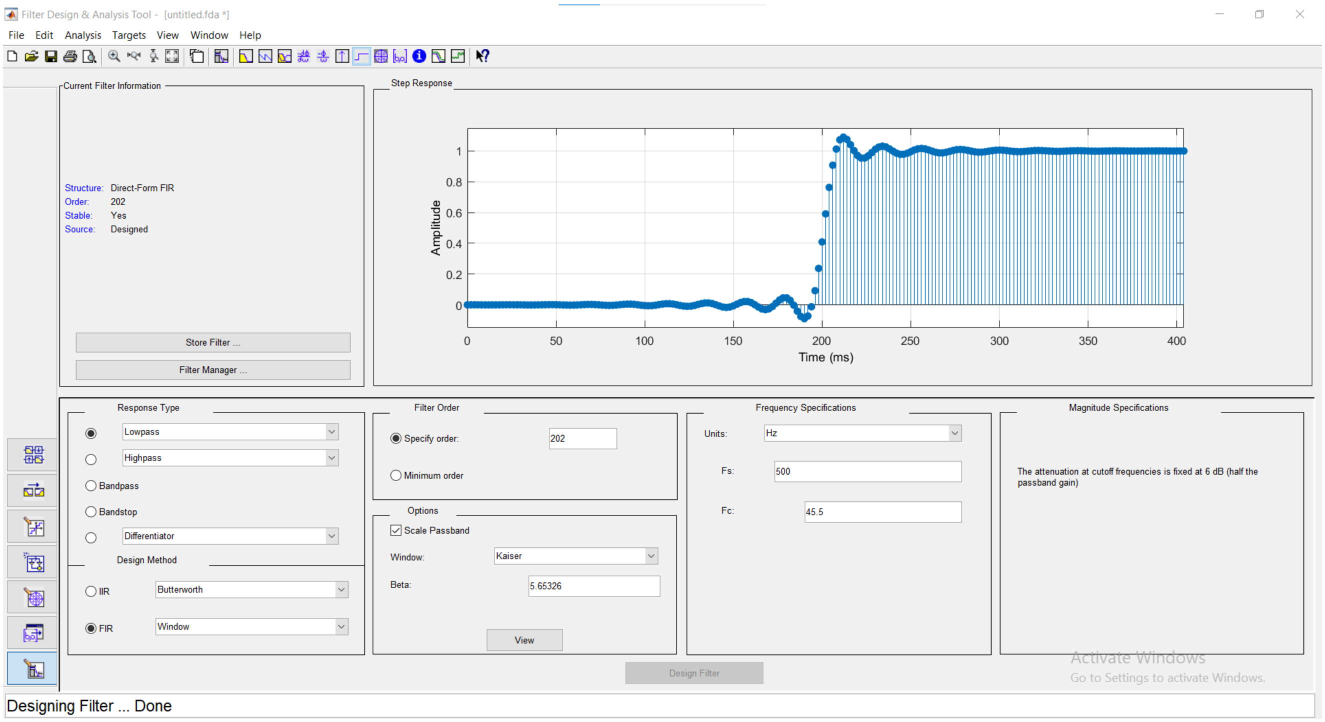

Figure-09: Step Response of Designed Filter

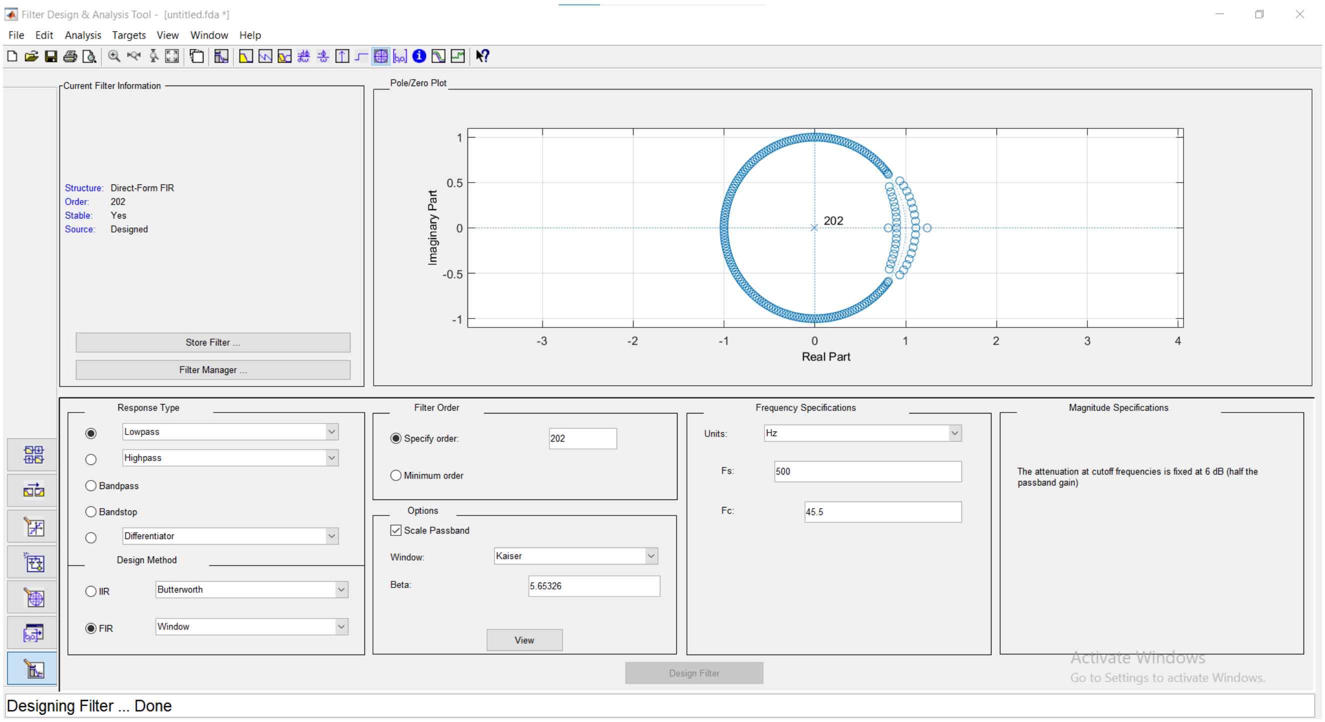

Figure-10: Poles & Zeros of Designed Filter

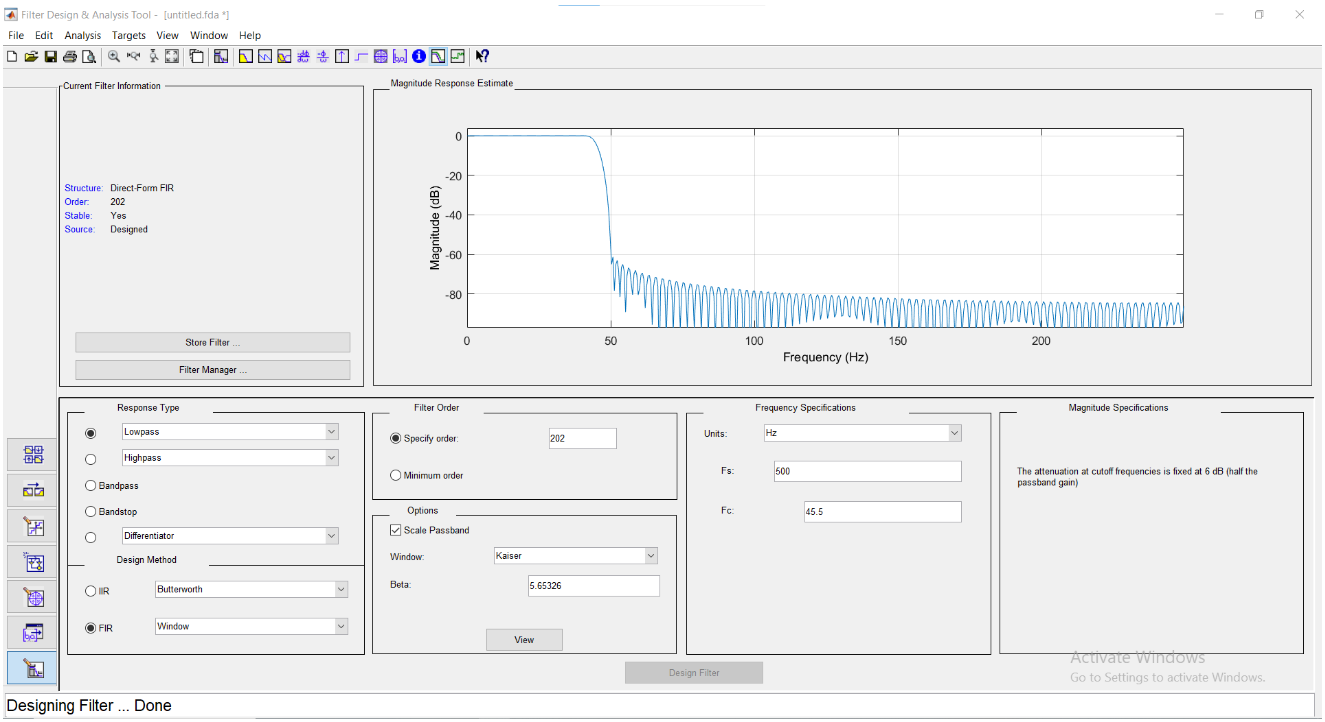

Figure-11: Magnitude Response Estimate of Designed Filter

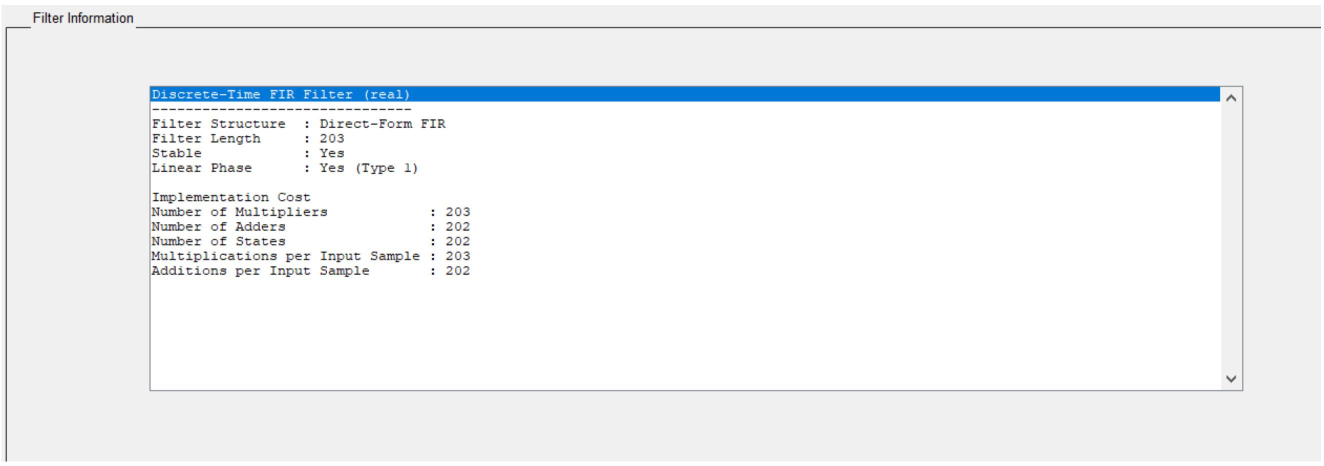

Figure-12: Specification of the Designed Filter

FILTER PERFORMANCE ANALYSIS:

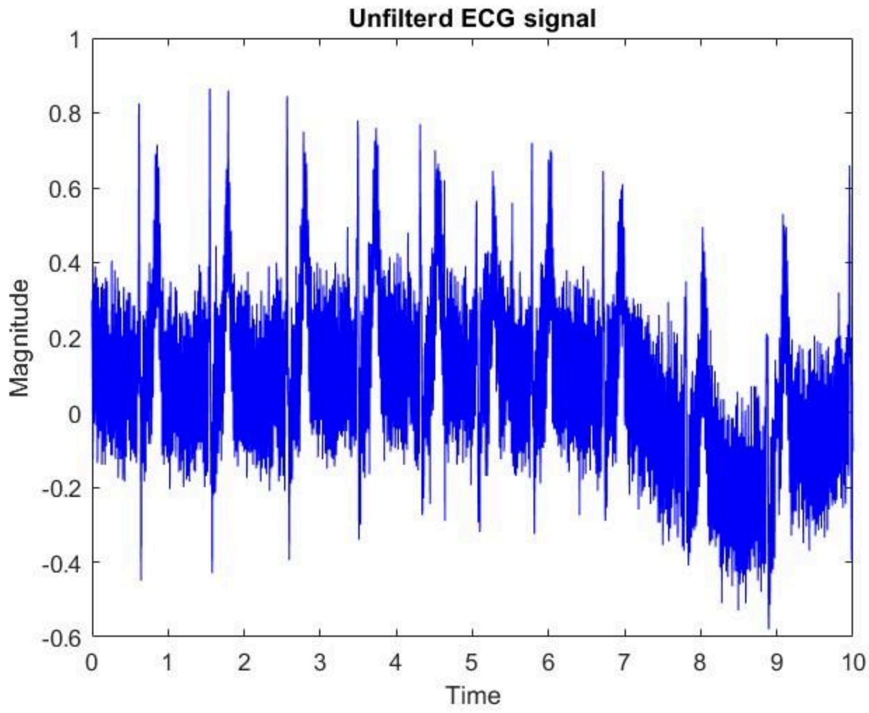

Figure-13: Provided unfiltered ECG signal (Time Domain)

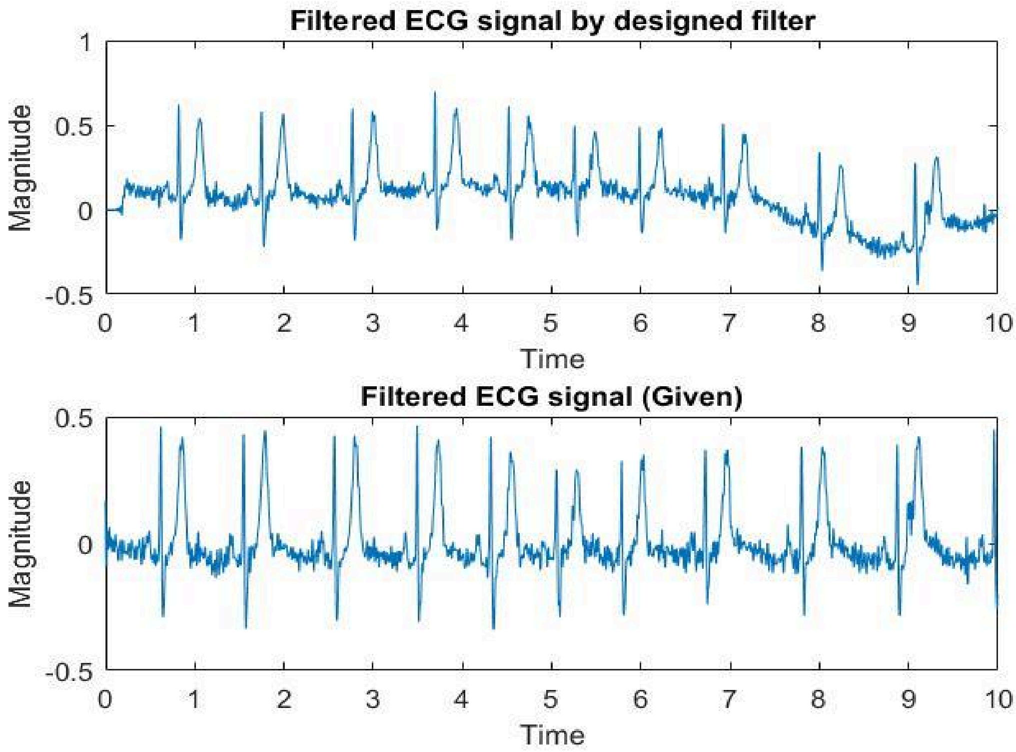

Figure-14: Filtered ECG signal by designed filter and Filtered ECG signal(Given)

Comment: If we compare between 2 subplot of Figure 14, we can see the filter ECG signal(Given) is clearer and less noisy. Because after filtration, noise gets removed from the given ECG signal.

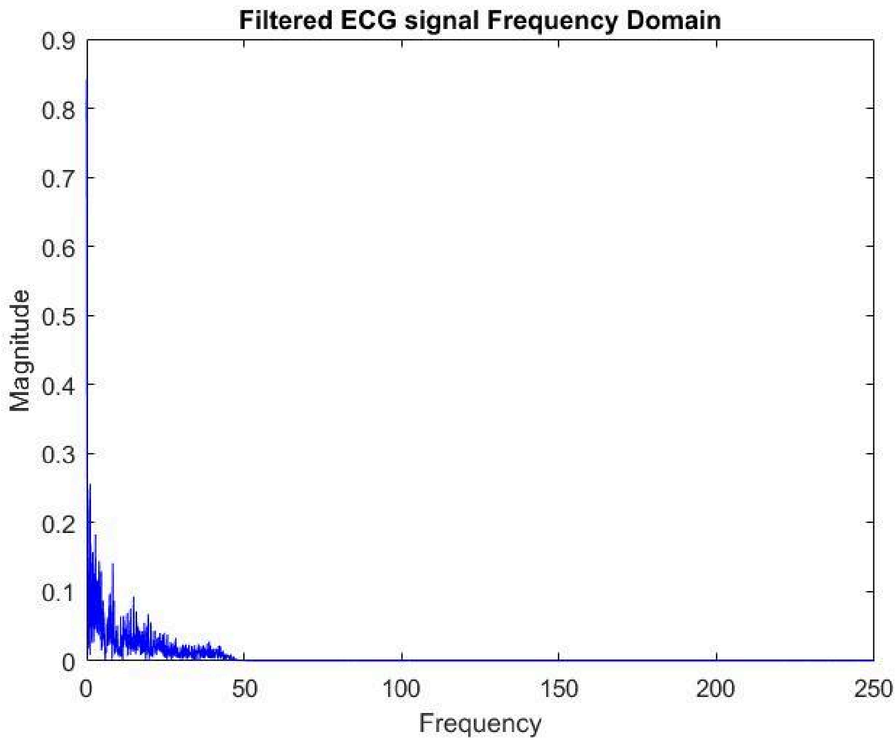

Figure-15: Filtered ECG signal Frequency Domain

Comment: We can observe that there is no noise in the signal in this figure. Thus, we may say that our distorted ECG signal has been properly filtered. We can see that the graph is much smoother, there are less noises, and there are no longer any unwelcome spikes in this image.

Improving the performance for future progress

1. Increase the filter order to improve performance. Higher-order filters generally provide steeper roll-off in the transition band, but be mindful of the increased complexity and potential for instability.
2. If we can reduce the number of memory usage, then the cost can be reduced, so designing our new filter will be better.
3. By designing new filters, we can reduce the noise. If the noise can be reduced, the results will be more accurate.

Limitations

In our FIR low pass filter design for ECG signals, we carefully balanced filter order, transition bandwidth, and Kaiser Window parameters. We optimized the shape parameter (beta) to achieve a delicate trade-off between main lobe width and side lobe levels. This ensured effective preservation of signal components and noise suppression. We meticulously managed passband ripple and stopband attenuation to accurately represent ECG signals while rejecting noise. Considering computational complexity and real-time processing, our design caters to diverse signal characteristics. Validation with representative ECG signals verified our filter's efficacy in meeting specific application requirements.

Conclusion

This design has taught us about several filtering ways. We can say the noise free signal and the filter signal have analogous characteristics. So we suppose that we've achieved the design outgrowth. This digital filter design will be really helpful further when we apply this knowledge to the engineering field. The FIR filter parameters were assumed and calculated. We selected the most efficient window while reducing the implementation cost & structure. The significance of this type of filter is similar to medical treatment. The actual information should be obtained through filters. Noise is an unavoidable part of a signal. Our FIR filter provides near-noise intended output at low implementation cost.

Appendix

Code:1

```
clc;
close all;
clear;
ECGsignal=load('Group_04.csv'); %loading all data
Fs=500;
t=ECGsignal(:,1); %taking time information
signal=ECGsignal(:,2);
figure(1)
plot(t,signal,'b') %plotting ECG data with time
title('Unfiltered ECG signal')
xlabel('Time')
ylabel('Magnitude')
grid on
axis tight
freq=signal/Fs;
l=length(freq);
nfft=2^nextpow2(l);
t1=abs(fft(freq,nfft));
freq1=Fs/2*linspace(0,1,nfft/2+1);
figure(2)
plot(freq1,t1(1:length(freq1)),'r');
title('Frequency domain of Unfiltered ECG signal')
xlabel('Frequency')
ylabel('Magnitude')
axis tight
grid on
```

Code:2

```
ECGsignal = load('Group_04.csv'); %Loading all data
load('filter.mat'); %Load Filter
Uf= ECGsignal(:,2); %taking signal information
t= ECGsignal(:,1); %taking time information from ECG

%Ploting Unfiltered ECG signal
figure(1)
plot(t,Uf)
title ('Unfiltered ECG signal')
xlabel('Time')
ylabel('Magnitude')
Filt=filter(Hd,Uf); % Unfilterd signal passing in filter

figure(2)
subplot(2,1,1)
plot(t,Filt)
title ('Filtered ECG signal by designed filter');
xlabel('Time')
ylabel('Magnitude')
Actu=ECGsignal(:,3);
subplot(2,1,2)
plot(t,Actu)
title ('Filtered ECG signal (Given)');
xlabel('Time')
ylabel('Magnitude')
figure(3)

Fs=500;
freq=Filt/Fs;
L=length(freq);
```

```
NEFT=2^nextpow2(L);
y1=abs(fft(freq,NEFT));
frequency=Fs/2*linspace(0,1,NEFT/2+1);
plot(frequency,y1(1:length(frequency)),'b');
title ('Filtered ECG signal Frequency Domain');
xlabel('Frequency')
ylabel('Magnitude')
```